What Is Claimed Is:

1. A VoIP system for voice communication over the internet, the VoIP system comprising:

a receiving system in which the receiving system is equipped with a voice data loss detection part for calculating a compressed voice data packet loss rate per designated time, a protocol header with a sequence number received over the internet, and a hold time decision-making part for deciding a current hold time per designated time using the voice data packet loss rate and generating hold time information packets; and

wherein, a transmission system for providing the compressed voice data packets, being equipped with a hold time display part for displaying the hold time based on the hold time information packets that are received over the internet.

2. The VoIP system of claim 1, wherein the hold time decision-making part has a first allowable value and a second allowable value lower than the first allowable value, and decides the current hold time by comparing the compressed voice data loss rate with a first allowable value, thus if the compressed voice data packet loss rate is larger than the first allowable value, the previous hold time increases by a designated time unit.

3. The VoIP system of claim 2, wherein the first allowable value is 5%, and the designated time unit is 1 second.

4. The VoIP system of claim 2, wherein the hold time decision-making part determines a current hold time by decreasing the previous hold time by a designated time unit if the compressed voice data loss rate is lower than a second allowable value.

5. The VoIP system of claim 4, wherein the second allowable value is 1%, and the designated time unit is 1 second.

6. The VoIP system of claim 1, wherein the hold time information packet is formatted to comprise an internet protocol header part, a service identifier part A for indicating the hold time information packet, a session ID number part B given to every telephone call, and a hold time part C for indicating a waiting time that commands the user to temporarily stop before staring

another sentence during a telephone call.

7. The VoIP system of claim 6, wherein the service identifier part A is 4 bytes, the session ID number part B is 3 bytes, and the hold time part C is 1 byte.

8. The VoIP system of claim 7, wherein if the hold time part C is 0000 0000, the hold time is 0 second, if the hold time part C is 0000 0001, the hold time is 1 second, if the hold time part C is 0000 0010, the hold time is 2 seconds, if the hold time part C is 0000 0100, the hold time is 3 seconds, if the hold time part C is 0000 1000, the hold time is 4 second, and if the hold time part C is 0001 0000, the hold time is 5 seconds, and the service identifier part A being 0000 0000 0000 0000 0000 0000 0000 0000.

9. The VoIP system of claim 1, wherein the designated time unit for calculating the packet loss rate is 30 seconds.

10. The VoIP system of claim 1, wherein the voice data loss detection part detects that the voice data loss is occurred as much as an increment if the sequence number increases more than 2 units, not increasing 1 unit gradually, while receiving the voice data packets.

11. A method for preventing data loss in the VoIP system, the method comprising the steps of:

transmitting compressed voice data in a form of a packet from the transmission system to the receiving system over the internet;

extracting the compressed voice data form the voice data packets received by the receiving system, and calculating a designated time unit of the compressed voice data packet loss rate;

deciding a current hold time per designated time unit by comparing the packet loss rate with a first allowable value and a second allowable value that is lower than the first allowable value in the receiving system; and,

transmitting the hold time information packet to the transmission system.

12. The method of claim 11, wherein a step is further included for displaying the hold time based on the hold time information received by the transmission system through a display means.

13. The method of claim 11, wherein the designated time unit is 30 seconds, the first allowable value is 5%, and the second allowable value is 1%.

14. The method of claim 11, wherein the total number T of the voice data packets and the voice data packet loss rate L are calculated applying the following algorithm:

$$T = M - N$$

$$L = (T - D)/T$$

wherein M is the maximum sequence number and N is the minimum sequence number among the voice data packets that are received per designated time unit.

15. The method of claim 11, wherein the hold time information packet is formatted to comprise an internet protocol header part, a service identifier part A for indicating the hold time information packet, a session ID number part B given to every telephone call, and a hold time part C for indicating a waiting time that commands the user to temporarily stop before staring another sentence during a telephone call.

16. The method of claim 15, wherein the service identifier part A is 4 bytes, the session ID number part B is 3 bytes, and the hold time part C is 1 byte.

17. The method of claim 16, wherein if the hold time part C is 0000 0000, the hold time is 0 second, if the hold time part C is 0000 0001, the hold time is 1 second, if the hold time part C is 0000 0010, the hold time is 2 seconds, if the hold time part C is 0000 0100, the hold time is 3 seconds, if the hold time part C is 0000 1000, the hold time is 4 second, and if the hold time part C is 0001 0000, the hold time is 5 seconds, and the service identifier part A being 0000 0000 0000 0000 0000 0000 0000 0000.

18. The method of claim 11, wherein the procedure, from deciding the current hold time to transmitting the current hold time information in a packet form to the transmission system,

comprises the steps of,

evaluating whether or not the compressed voice data loss rate is higher than the first allowable value,

based on the above evaluation, increasing the previous hold time by 1 time unit if the compressed voice data loss rate is higher than the first allowable value,

evaluating whether or not the increased hold time is higher than a designated maximum hold time,

based on the above evaluation, designating the increased hold time to be the current hold time if the increased hold time does not exceed the maximum value, and designating the maximum value to be the current hold time if the increased hold time exceeds the maximum value,

generating the hold time information packets based on the designated current hold time, and

transmitting the hole time information packets to the transmission system over the internet.

19. The method of claim 18, wherein the designated maximum value is 5 seconds, and the first time unit is 1 second.

20. The method of claim 11, wherein the procedure, from deciding the current hold time to transmitting the current hold time information in a packet form to the transmission system, comprises the steps of,

evaluating whether or not the compressed voice data loss rate is lower than the second allowable value,

based on the above evaluation, decreasing the previous hold time by 1 time unit if the compressed voice data loss rate is lower than the second allowable value,

evaluating whether or not the decreased hold time is lower than a designated minimum hold time,

based on the above evaluation, designating the decreased hold time to be the current hold time if the decreased hold time is not lower than the minimum value, and designating the minimum value to be the current hold time if the decreased hold time is lower than the minimum

value,

generating the hold time information packets based on the designated current hold time, and

transmitting the hold time information packets to the transmission system over the internet.

21. The method of claim 20, wherein the designated initial minimum value is 0 second, and the first time unit is 1 second.